# SETU ATA

## VoIP Adaptors with FXO, FXS and Multiple SIP Accounts

Internet Telephony offers intrinsic benefits of cost and flexibility. At the same time legacy telephony infrastructure and habits cannot be replaced overnight. People desire the best of both worlds - lower cost of VoIP and convenience of using existing telephony products and methods.

Matrix SETU ATA Range of Products is designed to meet this requirement of converting VoIP network to traditional telephony interfaces and vice-versa. It handles all the complexities of VoIP technology internally and provides simple telephone interfaces to make and receive calls.

Let Matrix SETU ATA be your bridge to the new world of IP Telephony!

MATRIX
TELECOM SOLUTIONS

# VoIP ATA With Unique Integration of Technology Features and Reliability

Matrix SETU ATA is a SIP based Analog Terminal Adaptor (ATA), it interfaces legacy telephone devices with IP-based networks. It is specially designed for SOHO users to offer them the advantages of low-tariff Internet Telephony for long distance calls, international calls and peer-to-peer calls. It can be used with any existing PBX providing users access to VoIP trunks. It can also be used in a stand-alone mode.

Matrix SETU ATA converts the voice traffic into data packets for transmission over the Internet. When a telephone number is dialed by a user, Matrix SETU ATA converts it into an IP call using the SIP protocol and initiates a call to the dialed number in any part of the globe. Using an appropriate VoIP service provider, long distance or inter-office call charges can be reduced significantly or eliminated through peer-to-peer calling on the IP network.

Making an outgoing call is as easy as from a normal telephone. Call progress tones like Dial Tone, Ring Back Tone and Busy Tone are fed to the caller as per the called number status. The FXS ports can make outgoing calls on a common or different SIP accounts. In addition, number based SIP account selection is provided to select the most economical SIP account for a given outgoing number.

An incoming call from a SIP account can be routed to any one or both FXS ports. All different CLIP protocols are supported so that the user can identify the caller before answering the call.

Once a call is established, features like Call Hold, Call Toggle, Call Transfer, Call Wait and Conference are supported to manage two calls from the same FXS port. Call forward in different conditions and Do Not Disturb is also provided.

The FXO port allows users to originate a call and terminate it at VoIP network. The routing logic used between FXO and VoIP helps the user to dial a destination PSTN number directly over VoIP network. Using FXO-VoIP routing logic two or more PBX can be networked to offer seamless connectivity between the PBXs

Matrix SETU ATA provides two Ethernet ports - one for WAN and the other for LAN. The user can connect his PC on the LAN port and browse the internet or check his emails while talking on VoIP calls.

Matrix SETU ATA can also be used with any PBX without changing its existing infrastructure. PBX users can make voice calls on IP to avail of the low-tariff of VoIP calls. The users can continue to make and receive calls without worrying on which network their calls are routed. Matrix SETU ATA is easy to install and operate. It can be configured using its built-in web pages served by the internal HTTP server.

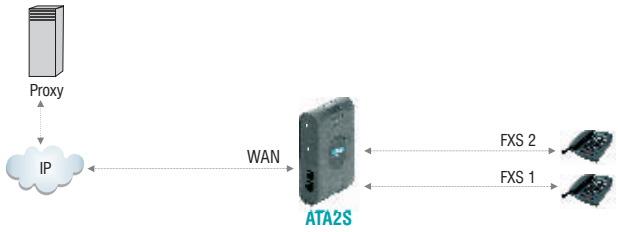
**SETU ATA1S**
1-FXS Port, 1-WAN Port,
1-LAN Port and 2-SIP Accounts

**SETU ATA2S**
2-FXS Ports, 1-WAN Port and
2-SIP Accounts

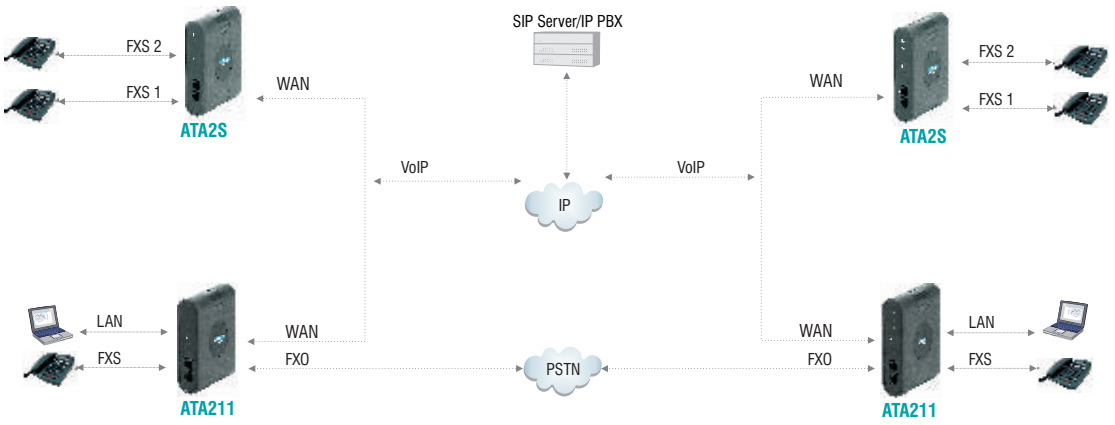
**SETU ATA211**
1-FXO Port, 1-FXS Port, 1-WAN Port,
1-LAN Port and 3-SIP Accounts

# PRODUCT APPLICATIONS

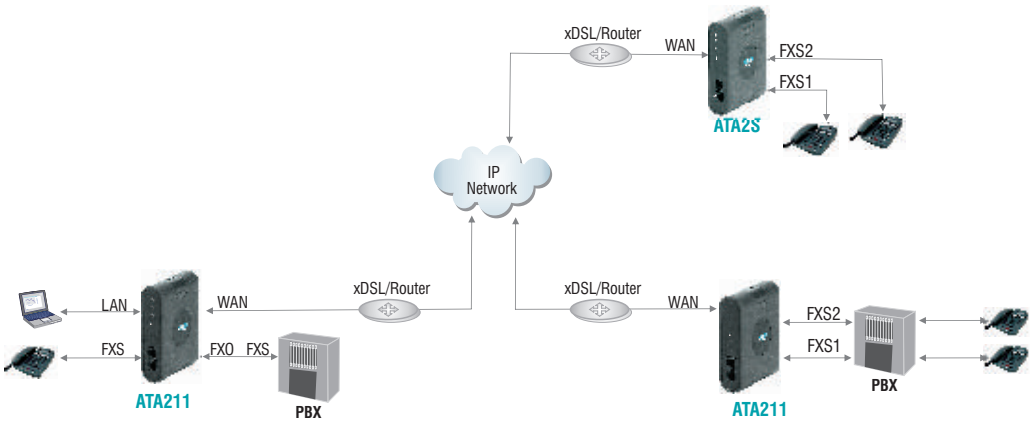
## RESIDENITAL APPLICATION




## BUSINESS APPLICATION




## PEER-TO-PEER CALLING APPLICATION



## KEY FEATURES

### Auto Configuration
SETU ATA can be configured automatically from a central location. The configuration details like Registrar Server Address, Authentication User ID, and User Password are stored in the central server. When user connects SETU ATA to the network, it automatically downloads its configuration using TFTP. This plug-n-play feature requires the user to enter only the server address provided by the service provider.

### Auto PSNT Fall back
PSTN can be interfaced to the SETU ATA using FXO Port. This port is used to dial out numbers to the PSTN Network. When Routing the calls from PSTN number to SIP trunk, it may happen that the Ethernet Link may go down or the SIP Account used is not registered. So the call will not be routed through SIP and you will get error tone. To avoid this you can use this feature Auto PSTN Fall Back through which the call will be automatically get routed through the alternate FXO Port.

### Automatic Number Translation
SETU ATA supports multiple port types, FXO, FXS and SIP. Whenever a number is dialed from any of these ports, gateway routes the call to the desired destination port as per the routing mechanism defined for that port. In certain cases, the dialed number string is not understood by the network through which the call is to be routed, so by using Automatic Number Translation the dialed number string is translated into a number that is understood by the network or ITSP to reach the desired destination port.

### Call Progress Tones and Rings
Matrix SETU ATA supports programmable tones and rings to match those of the country where it is installed.

### CLIP
SETU ATA allows users to program the FXS ports for any of the three CLIP protocols - DTMF, FSK ITU-T V.23 and FSK Bellcore 202A.

### Dial Number Table
Matrix SETU ATA provides a list of programmable numbers or part-numbers with the preferred SIP account for each entry. When the user dials a number, the SETU ATA finds the matching number using the “best-fit” logic. It then uses the SIP account given against this matching number to make that call. This ensures lowest cost for all the outgoing calls.

### Disconnect Signaling
When a call is released from the other side of the internet, the Matrix SETU ATA can propagate this call release on the FXS in the form of disconnect signal. The device senses this signal and frees the FXS port.

### Fax over IP (FoIP)
The user can send and receive Fax over SIP account, once a Fax machine is connected to SETU ATA. The SETU ATA supports FoIP using T.38 UDPTL and Pass Through.

### FXO
SETU ATA FXO port should be connected to the PSTN or PBX so that the user can make PSTN calls from the FXO port.

### Incoming Call Routing
Calls arriving from any SIP account can be routed to either one or both FXS ports.

### Jeeves (Web based Programming Tool)
Flexible and user friendly windows based software, Jeeves, helps in programming the features through web browser. This web based programming feature helps users to configure the SETU ATA from any part of the world once it is connected with the IP network.

### MAC Cloning
When replacing the existing hardware with other, you can simplify the installation process by copying the MAC Address of your existing PC. In such case, you do not need to delay the SETUp process by informing your service provider of newly installed equipment.

### Multi Stage Dialing
Multi-Stage dialing is useful for ATAs connected to a SIP Server used for networking PBX of multiple sites. The user can dial the entire number string, both the destination number and extension number of the destination PBX together. The ATA will split the string into two stages, and dial out the destination number first and on receiving the answering signal it dial extension number. This ensures hassle free access to PBX extension.

### Peer-to-Peer Calling
SETU ATA can make and receive calls from other VoIP users without any Registrar or Proxy server. Numbers and IP addresses can be assigned to the other VoIP users to provide direct access across the network. For Peer-to-Peer calling, SETU ATA provides two options - (i) Peer-to-Peer Number Dialing (ii) IP Address Dialing. Organizations having multiple locations like branch offices and factories can use this feature to provide direct dialing between these end-points.

### PIN Authentication
Incoming calls on FXS or FXO ports of the ATA can be restricted to a specific caller. The caller has to first prove his authentication before calling to ATA. This feature is used to avoid the possibility of malicious calls and to avoid misuse of its services.

### PPPoE
Matrix SETU ATA supports PPPoE client and hence can be used with any xDSL modems

### Router
Basic routing capabilities are provided so that LAN port packets can be transferred on WAN port. This allows the user to browse the internet and check his emails while making and receiving VoIP calls.

### SIP Accounts
Multiple SIP accounts can be programmed and each FXS user can be assigned one of the SIP accounts for outgoing calls. Dynamic allocation of SIP account is also possible using Dial Plan.

### Speech Volume Setting
SETU ATA allows user to set transmit and receive gain to improve the quality of speech.

### Speed Dialing
Frequently used numbers can be programmed in the internal phone book with 99 entries. The user can dial these numbers by using short codes in place of the complete, long numbers.

### Supplementary Services
SETU ATA supports supplementary service like Call Hold, Call Waiting, Call Toggle, Call Transfer, Call Forward, Conference, Caller ID, DND and Making another Call. These are the service provider dependant features.

### Surface Mount Technology (SMT)
The Surface Mount Technology is the current semi-conductor packaging technology that offers reduction in real estate resulting in less heat generation and low power consumption. This is in turn improves reliability.

### System Log
Syslog is one of the built in protocol used extensively for sending debug messages on IP network. This client/server protocol uses UDP as transport protocol for debugging process. Logging has several benefits which include troubleshooting, security and system administration. Debug messages are sent to remote server on IP network for finding and reducing the number of bugs or defects from a system.

## FEATURE LIST

- 100Rel/PRACK (RFC 3262)
- Answer Signaling
- Auto Configuration
- Auto PSTN Fallback
- Automatic Number Translation
- Called Party Number Table
- CLIP (DTMF, FSK-ITU-T V.23, Bellcore 202A)
- Comfort Noise Generation
- DHCP Client/ PPPoE
- Dialed Number Table
- Digest Authentication
- Disconnect Signaling
- Echo Cancellation
- Fax over IP-T.38 and Pass Through
- Flash Timer
- Forward Error Correction (FEC)
- Full Duplex Audio
- Incoming Call Routing
- LED Indications
- MAC Cloning
- Multiple Gateway Support
- Multistage Dialing
- Password Protection
- Peer-to-Peer Calling
- PIN Authentication
- Polarity Reversal
- PPPoE
- Programmable Call Progress Tones and Rings
- SIP over TCP
- Speech Volume Setting (Transmit and Receive)
- Speed Dialing
- STUN
- Supplementary Services
  - Call Forward Unconditionally
  - Call Forward on Busy
  - Call Forward on No Reply
  - Call Hold
  - Call Toggle
  - Call Waiting
  - Caller ID
  - Call Transfer-Blind
  - Call Transfer-Attended
  - Conference 3 Party
  - Do Not Disturb (DND)
  - Making Second Call
- Symmetric RTP
- Syslog Client
- Voice Activity Detection
- Web based GUI for Configuration

## MATRIX SETU ATA FEATURES LIST

| Features List | SETU ATA1S | SETU ATA2S | SETU ATA211 |
|---|---|---|---|
| 3-Party Conference | ● | ● | ● |
| Answer Signaling | ● | ● | ● |
| Auto Configuration | ● | ● | ● |
| Automatic Number Translation | ● | ● | ● |
| Auto PSTN Fall Back | ○ | ○ | ● |
| Called Party Number Table | ○ | ○ | ● |
| CLIP | ● | ● | ● |
| Comfort Noise Generation | ● | ● | ● |
| DHCP Client | ● | ● | ● |
| Dial Plan | ● | ● | ● |
| Digest Authentication | ● | ○ | ● |
| Disconnect Signaling | ● | ● | ● |
| Echo Cancellation | ● | ● | ● |
| Fax Over IP- T.38 and Pass Through | ● | ● | ● |
| Flash Timer | ● | ● | ● |
| Forward Error Correction | ● | ● | ● |
| Full Duplex Audio | ● | ● | ● |
| LED Indications | ● | ● | ● |
| Multiple Gateway | ● | ● | ● |
| Multi-Stage Dialling | ● | ● | ● |
| Password Protection | ● | ● | ● |
| Peer-to-Peer Calling | ● | ● | ● |
| PIN Authentication | ○ | ○ | ● |
| Phone Book | ● | ● | ● |
| Polarity Reversal | ● | ● | ● |
| PPPoE Client | ● | ● | ● |
| Call Progress Tone and Rings | ● | ● | ● |
| Speech Volume Settings | ● | ● | ● |
| Speech Dialing | ● | ● | ● |
| Supplementary Services | ● | ● | ● |
| STUN | ● | ● | ● |
| Symmetric RTP | ● | ● | ● |
| Syslog Client | ● | ○ | ● |
| Web based GUI for Configuration | ● | ● | ● |

● Available ○ Unavailable

## TECHNICAL SPECIFICATIONS

### VoIP

| | |
|---|---|
| VoIP Protocols | : SIP v2, SDP, RTP, RFC 2833 |
| Network Protocol | : IPv4, TCP, UDP, DHCP, SNTP, STUN, HTTP, PPPoE |
| SIP | : Multiple Accounts Out Bound Proxy Support, Display Name, User Name, Password, URL, Proxy URL, Registrar URL, Registrar Interval |
| NAT | : STUN and NAT Keep Alive |
| Voice CODECS | : G.711 A-Law, $\mu$-Law, G.723, G.729A, G.729B |
| Line Echo Cancellation | : G.168 with 8/16/32ms Tail Length |
| Call Progress Tones | : Dial Tone, Ring Back Tone, Busy Tone, Error Tone |
| Voice | : Dynamic Jitter Buffer (Adaptive), Comfort Noise Generation and Voice Activity Detection |
| Fax | : T.38 and Pass Through |
| Quality of Service | : Layer 3 DIFFServ and TOS |
| Data Network | : WAN Port (RJ45), Auto MDIX 10/100 BaseT, LAN Port (RJ45), Auto MDIX 10/100 BaseT |
| Security | : Password Protected Administration |

### FXS Port

| | |
|---|---|
| Connection | : RJ11 |
| Off Hook Impedance | : 600 Ω |
| Loop Limit | : 270 Ω (Max) Excluding Telephone Set |
| Loop Feed | : 39mA (Max) |
| Ringing Voltage | : 55Vrms @25Hz, 3REN |
| Pulse Dialing | : 10 PPS and 20PPS @ 1:2, 2:3 and 1:1 |
| DTMF Dialing and Reception | : ITUT Q.23 and Q.24 |
| Caller ID Presentation (CLIP) | : DTMF, FSK ITU-T V.23 and FSK Bellcore 202A |
| Call Maturity | : Polarity Reversal |
| Protection | : Solid State (Over Voltage and Over Current) built-in Secondary Protection |

### FXO Port

| | |
|---|---|
| Connection | : RJ11 |
| Off Hook Line Impedance | : 600 Ω |
| Loop Limit | : 1500 Ω |
| Pulse Dialling | : 10 PPS and 20 PPS @ 1:2, 2:3 and 1:1 |
| DTMF Dialling and Reception | : ITU-T Q.23 and Q.24 |
| CLI Reception | : DTMF, FSK ITU-T V.23 and FSK Bellcore 202A |
| Call Maturity | : Polarity Reversal |
| Protection | : Solid state (Over Voltage and over current) built-in secondary Protection |

### Power Supply

| | |
|---|---|
| Input | : 12VDC @1.25A through External Adaptor (90-265VAC, 47-63Hz) |
| Connector | : DC Power Jack |

### Power Consumption

| | |
|---|---|
| SETU ATA1S | : 5W Approx. |
| SETU ATA2S | : 5W Approx. |
| SETU ATA211 | : 5W Approx. |

### Mechanical

| | |
|---|---|
| Dimensions (WxHxD) | |
| SETU ATA1S | : 7.9x10.5x2.7cm (3.1"x4.1"x1.1") |
| SETU ATA2S | : 7.9x10.5x2.7cm (3.1"x4.1"x1.1") |
| SETU ATA211 | : 7.9x10.5x2.7cm (3.1"x4.1"x1.1") |
| Unit Weight | |
| SETU ATA1S | : 0.45Kgs (1.10lbs) Approx. |
| SETU ATA2S | : 0.45Kgs (1.10lbs) Approx. |
| SETU ATA211 | : 0.45Kgs (1.10lbs) Approx. |
| Shipping Weight | |
| SETU ATA1S | : 1.00Kgs (2.20lbs) Approx. |
| SETU ATA2S | : 1.00Kgs (2.20lbs) Approx. |
| SETU ATA211 | : 1.00Kgs (2.20lbs) Approx. |
| Material | |
| SETU ATA1S | : ABS Plastic |
| SETU ATA2S | : ABS Plastic |
| SETU ATA211 | : ABS Plastic |

### Installation Mounting

| | |
|---|---|
| SETU ATA1S | : Wall and Table-Top |
| SETU ATA2S | : Wall and Table-Top |
| SETU ATA211 | : Wall and Table-Top |

### Environmental

| | |
|---|---|
| Operating Temperature | : -10°C to +50°C (-14°F to +122°F) |
| Storage Temperature | : -40°C to +85°C (-40°F to +185°F) |
| Operating Humidity | : 5-95% RH (Non-Condensing) |
| Storage Humidity | : 0-95% RH (Non-Condensing) at 40°C |

## SYSTEM CAPACITY AND RESOURCES

| Hardware | SETU ATA1S | SETU ATA2S | SETU ATA211 |
|---|---|---|---|
| FXS Ports | 1 | 2 | 1 |
| FXO Ports | - | - | 1 |
| LAN Port | 1 | - | 1 |
| WAN Port | 1 | 1 | 1 |
| DC Power Jack | 1 | 1 | 1 |

## COMPLIANCES

| EMI/EMC | |
|---|---|
| Conducted Emission | : CISPR 22 Class A |
| Radiated Emission | : CISPR 22 Class A |
| Harmonic Current Emission | : IEC 61000-3-2 |
| Voltage Flicker | : IEC 61000-3-3 |
| Electro-static Discharge | : IEC 61000-4-2 |
| Radiated Susceptibility | : IEC 61000-4-3 |
| Electrical Fast Transient | : IEC 61000-4-4 |
| Surge | : IEC 61000-4-5 |
| Conducted Immunity | : IEC 61000-4-6 |
| Power Frequency Magnetic Field | : IEC 61000-4-8 |
| Voltage Interruption & Dips | : IEC 61000-4-11 |

| FCC | |
|---|---|
| Conducted Emission | : FCC Part 15 Sub Part B Class A |
| Radiated Emission | : FCC Part 15 Sub Part B Class A |

| EC Directives |
|---|
| R&TTE 1999/5/EC |
| LVD 73/23/EEC |
| EMC 89/336EEC |

| Safety | |
|---|---|
| IEC 60950-1 | : 2001 First Edition |

## VoIP PRODUCTS FROM MATRIX

| | |
|---|---|
| SETU ATA1S | SIP based Analog Telephone Adaptor with 1 FXS Port and 2 Ethernet Ports |
| SETU ATA2S | SIP based Analog Telephone Adaptor with 2 FXS Ports and 2 Ethernet Ports |
| SETU ATA211 | SIP based Analog Telephone Adaptor with 1 FXO, 1 FXS Port and 2 Ethernet Ports |
| SETU VFX44L | SIP based VoIP Gateway with 4 FXS Ports, 1 FXO (PSTN Pass-Through) and 1 Ethernet Port |
| SETU VFX88L | SIP based VoIP Gateway with 8 FXS Ports, 1 FXO (PSTN Pass-Through) and 1 Ethernet Port |
| SETU VP236S | Executive IP-Phone with 2 Lines x 24 Characters LCD Display |
| SETU VP236SE | Executive IP-Phone with 2 Lines x 24 Characters LCD Display and PoE |
| SETU VP248SE | Executive IP-Phone with 2 Lines x 24 Characters LCD Display and PoE |
| SETU VP248PE | Executive IP-Phone with 6 Lines x 24 Characters LCD Display and PoE |
| SETU VGFX8422 | Multi-Port VoIP Gateway with 8 VoIP Channels, 4 GSM Channels, 2 FXO and 2 FXS Ports |
| SETU VGFX8440 | Multi-Port VoIP Gateway with 8 VoIP Channels, 4 GSM Channels and 4 FXO Ports |
| SETU VGFX8404 | Multi-Port VoIP Gateway with 8 VoIP Channels, 4 GSM Channels and 4 FXS Ports |
| SAPEX IPXP250 | All-Integrated IP PBX for 250 Users |
| SAPEX IPXP500 | All-Integrated IP PBX for 500 Users |

## ABOUT MATRIX

An ISO 9001 Company, Matrix is a leader in the VoIP, GSM, Key Phone System and PBX market. An innovative, technology driven and customer focused organization; the company is committed to keep pace with revolutions in the telecom industry. This has resulted in bringing forth cutting edge products like VoIP Phones, VoIP Gateways, VoIP ATA, GSM FCT, GSM Gateways, SMB PBX, Enterprise PBX, Hotel PBX, Voice Messaging Products , Communication Security Products and PLCC Switches. With over 1,500,000 line units installed and growing by over 1500 line units per day, the installed base of Matrix connects over 15,000,000 calls everyday. Thus, Matrix has gained the trust and admiration of users representing the entire spectrum of industries. Matrix has won many awards for its innovative products.

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*Due to continuous technology upgradations, product specifications are subject to change without notice.*